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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
08/824,496	03/14/1997	J. CARL COOPER	JCC396A	8681

7590 12/31/2002

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ART UNIT	**PAPER NUMBER**
2644	

DATE MAILED: 12/31/2002

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BEFORE THE BOARD OF PATENT APPEALS

AND INTERFERENCES

Paper No. 36

Application Number: 08/824,496

Filing Date: March 14, 1997

Appellant(s): J. Carl Cooper

J. Carl Cooper

<u>For Appellant</u>

EXAMINER'S ANSWER

This is in response to the appeal brief filed October 17, 2002.

(1) Real Party in Interest

A statement identifying the real party in interest is contained in the brief.

(2) Related Appeals and Interferences

A statement identifying the related appeals and interferences which will directly affect or be directly affected by or have a bearing on the decision in the pending appeal is contained in the brief.

(3) Status of Claims

The statement of the status of the claims contained in the brief is correct.

(4) Status of Amendments After Final

The appellant's statement of the status of amendments after final rejection contained in the brief is correct.

(5) Summary of Invention

The summary of invention contained in the brief is correct.

(6) Issues

The appellant's statement of the issues in the brief is correct.

(7) Grouping of Claims

Appellant's brief includes a statement that claims 1 to 38 and 40 to 53 do not stand or fall together and provides reasons as set forth in 37 CFR 1.192(c)(7) and (c)(8).

(8) Claims Appealed

The copy of the appealed claims contained in the Appendix to the brief is correct.

(9) Prior Art of Record

The following is a listing of the prior art of record relied upon in the rejection of claims under appeal.

5,636,323	UMEMOTO	6-1997
4,268,727	AGRAWAL	5-1981

Kuo, Sen M. "Active Noise Control Systems", 1996 (Preface by authors dated October, 1995), pp 35-36.

0170298	TANNO	10-1983
4,025,724	DAVIDSON	5-1977

(10) Grounds of Rejection

The following ground(s) of rejection are applicable to the appealed claims:

1. Claims 1 and 4 to 19 are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

2. Claims 1 and 19 requires that both delay and gain be automatically adjusted, whereas the parent claim requires that either delay or gain is human operator adjustable. Claims 1 and 19 therefore directly contradicts claim 1, making claims 19 and 1 indefinite.

3. (e) the invention was described in a patent granted on an application for patent by another filed in the United States before the invention thereof by the applicant for patent, or on an international application by another who has fulfilled the requirements of paragraphs (1), (2), and (4) of section 371© of this title before the invention thereof by the applicant for patent.

The changes made to 35 U.S.C. 102(e) by the American Inventors Protection Act of 1999 (AIPA) do not apply to the examination of this application as the application being examined was not (1) filed on or after November 29, 2000, or (2) voluntarily published under 35 U.S.C. 122(b). Therefore, this application is examined under 35 U.S.C. 102(e) prior to the amendment by the AIPA (pre-AIPA 35 U.S.C. 102(e)).

4. Claims 2, 3-7, 18-27, 29 to 31, 37 and 28 are rejected under 35 U.S.C. 102(e) as being anticipated by Umemoto (refereed to as "Umemoto" hereafter; note that the earlier filed PCT of this case was published on 8/4/94).

Claim 3 is exemplary for the basic components: Talent signal (coming in on RS); feedback signal (via acoustic feedback path EC, which also contributes locally generated speech intended for microphone 14--see Umemoto, col. 4, lines 29-36); variable delay and variable gain is

adaptive filter ADF (see fig. 3, each tap output represents a delay, the gain of which may be independent set) ADF is automatically updated according to the error signal ESS (which ideally contains only the locally generated speech which is input to microphone 14, less the feedback signal from the far end which propagates from loudspeaker 13 to microphone 14) which is equivalent to applicant's mix minus signal; combining circuit is junction 32; and the feedback signal is, of course, provided without a variable delay circuit.

For claims 4-7, 23-27, etc., note that the coefficients are updated according to ESS, but ESS=Feedback Signal - ADFOutput. Consequently, it is true that both the delay(s) and gain(s) of the taps are a function of ESS (the mix minus signal), but the mix minus signal is just a function of (at least) the Feedback signal. Thus the delay(s) and gain(s) may be said the be function of the Feedback signal. Hence the broad language of these claims is met.

5. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless -
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

6. Claims 2-7, 10, 11 and 16 to 18 are rejected under 35 U.S.C. 102(b) as being anticipated by Agrawal et al ("Agrawal"), using a similar analysis to the used above.

In this reference, an explicit correlation using a correlator is disclosed.

7. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made

to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

8. Claims 8 to 17, 28 and 32 to 36 are rejected under 35 U.S.C. 103(a) as being unpatentable over Umemoto considered with the publication to Kuo et al (Active Noise Control Systems, pp 35-36).

As to claims 8 to 17, first, not that a correlator or use of a correlator is not recited, only that concept of a correlation between two signals.

Note that there is always inherently a correlation (be it positive, zero, or negative) between any two signals. As is well known in adaptive filtering, as mentioned by Kuo in their discussion of the Correlation LMS Algorithm, as the filter converges, the correlation between the error signal and the signal input to the filter decrease.

Thus in claim 8, for example, we already know that the delay and gain are adjusted according to the error signal ESS ("mix minus"); the magnitude of the error signal and its correlation are generally proportional, hence, the delay and gain at any iteration are determined (at least in part) by the correlation of filter input ("talent signal") and error signal ESS ("mix minus").

The inherent correlation being spoken of (this is distinct from the algorithm Kuo discuss, which is not itself being invoked in this discussion) at any instant exists between the feedback signal and the talent signal at any phase of the talent signal. There is an inherent correlation associated with the feedback signal and the undelayed signal, as well as between the feedback signal and the net filter output, which, of course, is delayed version of the talent signal, and so forth.

If the relation or correlation of feedback signal and undelayed talent signal at any iteration is x, the correlation between the feedback signal and the filter output is fixed according to the current tap weights.

Thus if the error signal is responsive to the correlation between the feedback signal and the filter output, it is also inherently be responsive to whatever the instantaneous correlation is between the feedback signal and the undelayed talent signal, taking the tap weights (and consequent delay introduced to the talent signal between the filter input and output) into account.

Hence the delay(s) and gain(s) are inherently responsive to the correlation between the feedback signal and the talent signal (RS in Umemoto) or a delayed version of it from the filter ADF.

Hence claims 8-17 are met. Note that the cancellation signal is the filter output (re claims 16, 17).

As to claim 20, it appears to be met by the foregoing discussion. The recitation "providing a cancellation signal of a known level in response to said delayed talent signal" is considered as met since the level of the filter output is known or knowable by its input and the instantaneous tap weight values.

Claim 21 is similar to the various preceding claims; note that "providing said mix minus signal in response to said feedback signal and said cancellation signa" is inherently met since the error signal ESS is the difference of the feedback signal and the filter output, which is the cancellation signal.

Claim 22 appears to be met in view of all of the preceding discussion.

Claims 23 to 38 also appear to be met. The word "comparison" in claim 38 is considered to be conceptually equivalent to "correlation", since correlation is just a type of comparison, and comparison is a type of correlation.

9. Claims 1-17, 40, 41 and 43-47 are rejected under 35 U.S.C. 103(a) as being unpatentable over Tanno
considered with Davidson and Kuo.

Regarding claim 1, Tanno shows a simple acoustic echo canceler. A person speaking into M1 would, without D1 and T2, hear his voice in a feedback loop, as his voice would be aurally broadcast by speaker S1, picked up after a short acoustic delay in air by M2, and rebroadcast by speaker S2, to be picked up again by M1, and so on, creating echo and/or howlaround.

The limitations of claim 1 are read as follows: person (talent) speaks into mic M1; cancellation circuit has delay D1 (variable), cancellation signal is the output of delay D1; feedback signal is the signal entering M2 which was broadcast by S1; combining circuit is T2; the result is a mix minus signal in the sense that the mixed signal is the sound from M2, which includes the user's voice at M2 plus background sound, plus the acoustic feedback of the voice from the user at M1, and the voice from the user at M1 is subtracted at T2 from the mixed signal. Also, the feedback signal is "applied without the use of a variable delay circuit"/ What Tanno does not show is that the delay D1 is user adjustable, and Tanno does not show a variable gain circuit.

In a similar field, interference cancellation, Davidson shows the use of manually adjustable delays to delay a signal sufficiently to allow it to cancel an acoustically delayed version of itself.

Since Tanno gives no guidance as to how the delays are adjusted, it would have been obvious to use any well known technique in the art, such as that of Davidson, which is manual or human operator adjustment.

Not that Davidson recognizes, by providing plural delays, that both direct and indirect paths may be taken, and thus plural delays are needed to model the acoustic path. Similarly , in Tanno, it would have been obvious to provide any number of manually adjustable delay paths for D1 and D2, at the particular use may require.

Regarding the need for variable gain, Tanno desires, for example, that the signal from M1 at point Q be canceled by the injection of a delayed version of itself at T2. One of ordinary skilled would realize that in order for this to happen, the delayed signal from M1 via D1 must be subtracted from the signal from M2 (which would contain an acoustically delayed version of the signal from M1 which was broadcast by speaker S1).

If we designate the gain function from S1 to M2 as GS1-M2, and whatever gain with which the output of D1 is injected into the T2 combining junction as GinjT2, and the signals from M1, etc., simply as M1, etc., (using the part name as the signal name for simplicity), we can write a simple expression for the signal at point Q, which we'll just call Q:

Q=(M1)(A1)(GS1-M2)-(M1)(GinjT2)

Not that this is true when D1 is properly adjusted so its delay and the acoustic path delay are the same.

Since we are only discussing the signal M, and we want Q to be zero for M1, we need to set Q=0 in the above equation and solve for GinjT2, noting that in general, A1 is likely to be variable via a volume control, and the acoustic transfer function is likely to be variable due to environmental changes.

GinjT2=(A1)(GS1-M2)

Unless it can be guaranteed, and known ahead of time that A1 and GS1-M2 will always be constant (this is very unlikely to be the case), then clearly a variable gain adjustment must be provided for GinjT2 and it would have been obvious to do so in Tanno, such as by placing a potentiometer at the output of D1, if D1 is a single delay, or multiple potentiometers if D1 is implemented according to Davidson with plural delays to account for plural paths (both direct and different echo or indirect paths).

The Tanno, Davidson, Kuo combination thus far described meets claims 1, 40 43 and 46; as to claim 41, it would have been obvious for operator using the combination to adjust the device in any manner desired, such as to fully attenuate the echo, or only partially, if even only to hear what partial echo cancellation sound like, even if only once.

As to claim 44, since the operator listens to the result which is essentially the error signal for M1 at point Q (its emitted by S2), and would adjust delay and gain in response thereto, it certainly would be true that "the amount of said gain responsive to said mix minus signal....."

As to claims 45 and 47, they are treated as 41.

Continuing the discussion with claim 2, which is an independent claim, it requires that the delay and /or gain responsive to said mix minus signals, etc., This limitation is met for the same reasons as discussed with respect to claim 44.

Claims 4-17 are rejected using reasoning similar to that given in the rejection supra using Umemoto and Kuo.

10. Claims 18, 42 and 48-53 are rejected under 35 U.S.C. 103(a) as being unpatentable over Tanno, Davidson, Kuo combining as given above, further considered with either pf Agrawal or Umemoto or alternately, either of Agrawal or Umemoto considered with Davidson.

While the Tanno, Davidson, Kuo combination is a primitive working embodiment, manually adjustable, and no doubt suitable for certain situations without rapidly changing conditions, it would have been obvious to provide an alternate automatic coefficient update mechanism which could be switched in when the user desired to have automatic delay. Such a combination would then both adjustable and automatically adjustable delay and gain.

Such a combination is in fact, merely an arrangement to alternately switch between known (i.e., unpatentable) combination with no new or unexpected result. See Duplan Corp. V. Deering Milliken, 197 USPQ 342 (#97).

A similar line of reasoning applies to the alternate rejection, in which the fully automatically updated adaptive filter reference would be provided with an alternative, manually operated arrangement. Besides the reasoning of Duplan Corp, provision of manual override controls in any (automatically) adaptive system for purposes of manual control when the coefficient update algorithm misbehaves (i.e., coefficients diverge, such as when a two sinusoidal signal in the channel, with insufficient noise).

Either of these combinations would read on claim 42, for example, when the operator sets the delay and/or gain in manual mode, and then decides to summarily switch to automatic mode. Presumably, the adaptive filter, automatically operated, would converge (within 10 times the filter length) rapidly to "the expected amount", as recited in claim 42. Claims 48-50, 52 and 53 read similarly. Note that these claimed do not recite-automatically initiating the change of operational mode from manual to automatic. They only recite a value which had been set

manually is (ostensibly) later change by an automatic process, which reads on an LMS adaptive filter converging, the manner of initiation or switching from manual to automatic mode not being limited by the claims.

Claim 51 is also considered obvious for the same reasons as claim 45, for example.

(11) Response to Argument

On page 7, line 14 to 23, the applicant has argued that "the examiner has not provided any support or reasoning which would cause the person of ordinary skill in the industry to which the present invention pertains (set forth for example in the Field of the Invention '....audibly communicating with remotely located actors and reporters in radio and television system') to look to the presently cited prior art which pertains to different fields of art (for example point to point communications such as the television system of Agrawal and the mobile telephone system of Umemoto), to achieve the claimed invention". The appellant's argument is not persuasive because even though Agrawal or Umemoto systems may be different from the appellant's claimed invention, as described above Agrawal and Umemoto references read on the claims.

It is the claims which are the measure of the invention, not the disclosure.

On page 8, lines 11 to 22, the appellant has argued that "the term 'program signal' would be known to the person of ordinary skill to pertain to a mixture of electronic signals including the talent signal in recorded or broadcast production of television and radio like programs". The appellant's argument is not persuasive because the claims do not recite such limitations, and certainly, the term "program signal" in its broadest reasonable interpretation need not be limited to what Appellant has argued; if Appellant wishes for the term to be limited to something specific, the details must be recited to be given weight in the case of so broad a term as "program signal".

On page 9, lines 6 to 17, the appellant has argued that "the invention may be practiced with automatic adjustment to track changes of delay and/or gain occurring in the feedback signal as well as manual adjustment to provide a small amount of talent's voice in the mix minus signal (i.e. lees than full cancellation) as described at the bottom paragraph of page 13 and the middle paragraph of page 20". The appellant's argument is not persuasive because the bottom paraph of page 13 and the middle paragraph of page 20 still does not disclose using both manual and automatic adjustment together, simulataneously, which is the Examiner's interpretation of using automatic delay ***and*** gain.. It appears that it only discloses using automatic ***or*** manual adjustment.

On page 10, lines 1 to 11, the appellant has argued that "the person of ordinary skill in the art would know that none of the signals which occur in telephone systems of the typed described

on the present prior art are 'mix minus signals' (a special signal including program audio with the talent's voice removed or reduced) as that term is used in the claims". The appellant's argument is not persuasive because it is unclear what the appellant means by "a special signal". Also, the claims do not recite such limitations. Furthermore, the broadest reasonable interpretation of "mix-inus" signal is some kind of mixed signal less or minus something else. For example, the signal in Umemoto entering the error junction 32 has the signal EC (echoed version of RS) plus the locally generated speech signal captured by microphone 14 (along with EC, of course). The signal exiting error junction 14, is therefore a mixed signal (EC plus locally generated speech) minus another signal (a delayed version of the talent or RS signal).

On page 10, lines 12 to 22, the appellant has argued that "examiner refers to Umemoto's EC as the feedback signal, but RS is not mixed with other signals to provide EC, thus while RS may come from a human voice, RS does not fall within the meaning of, and thus can not be the 'talent signal' of the claims". The appellant's argument is not persuasive because the claims do not recite that the talent signal is a signal intended to be mixed with other electronic signals. Furthermore, it is in fact mixed with another signal (the local speech signal) before entering microphone 14.

On page 10, line 23 to page 11, line 4, the appellant has argued that "Umemoto's EC is an acoustic signal, not an electronic signal. EC is also not a mixture of electronic signals including the talent signal (which the examiner equates to RS) and it is not recorded or broadcast (program signal)." The appellant's argument is not persuasive because as described above, the limitations that the appellant is arguing do not appear in the claims.

On page 11, lines 5 to 22, the appellant has argued that "the claim term 'mix minus' would not be considered by a person of ordinary skill to be equivalent to Umemoto's ESS signal. The mix minus signal of the claims is the approximation of the program signal without the talent signal". The appellant's argument is not persuasive because it is not clear what the appellant means by "the approximation of the program signal". What program signal is it being referred to? Also, the claims do not recite that the mix minus signal is "an approximation of the program signal". Furthermore, the talent signal (RS) in Umemoto, for example, is mixed with another signal, the locally generated speech signal, which may certainly be referred to as a "program signal", just as much as any other speech or music signal may be considered as a program signal. ***What is or is not a "program" signal is a matter of intended use rather than a structural limitation which could distinguish one speech signal from another.***

On page 12, lines 1 to 17, the appellant has argued that "the claimed feedback signal is an electronic signal, whereas the signals entering M2 is an acoustic signal". The appellant's argument is not persuasive because the claims do not recite that the feedback signal is an electronic signal. Also, the feedback signal of Tanno is an electronic signal because it is produced by an electronic system. Also, the appellant has argued that "if the feedback signal is considered to be the electronic signal out of M2, it is not delayed as called for in the rejected

claims". The appellant's argument is not persuasive because the claims do not recite that the feedback signal is delayed. In addition, the appellant has argued that "one of ordinary skill in the art would know that attenuating circuit would operate to adjust the level of the signal from M2, not to combine the signal from D1 with the signal from M2". Th appellant's argument is not persuasive because the claims claimed "said feedback signal and said cancellation signal being applied to a combining circuit....". Therefore, T2, which is an attenuation circuit, reads on for combining the feedback signal (input of M2 and the cancellation signal (output of D1).

On page 12, lines 18 to 22, the appellant has requested explanation of the effective date of Kuo reference, since the present application receives priority from provisional application 60/013,545 filed 03/14/96. Kuo reference is pages from a text book called "Active Noise Control Systems" which contains preface, pp xi to xiv, dated October 1995 (a copy has been attached with Examiner's Answer). Given above, Kuo reference could be still applied. This shows that others, i.e., the two authors of the book, were in possession of the knowledge of the book earlier than the broadly stated publication date of the book, 1996.

On page 13, lines 1 to 11, the appellant has argued that "claim 1 does not recite either delay or gain adjustment as the examiner states in the rejection, Claim 1 thus could cover a system where the delay and gain are or the gain is operator adjusted, or the gain is operator adjusted or where both the delay and gain are operator adjusted". The appellant's argument is not persuasive because as described above, it is inconsistent to use both manual and automatic adjustment for gain and delay.

On page 14, lines 6 to 8, the appellant has argued that "by contrast, in applicant's invention of the rejected claims, a portion of the talent signal is combined with the feedback (received) signal to cancel the delayed talent portion which is present in the feedback signal". The appellant's argument is not persuasive because the claims claimed that "the feedback signal and said cancellation signal being applied to a combining circuit to provide said mix minus signal with said feedback signal being applied without the use of a variable delay circuit". As earlier remarked in regard to Umemoto, for example, the output of error junction 32 contains the local speech ("program signal") entering 14 plus the talent signal (RS), minus a delayed version of the talent signal (RS) from ADF 31.

On page 14, lines 9 to 24, the appellant has argued that "applicant's claimed feedback signal is a combination of program material and talent signal and the invention operates to provide a mix minus signal in response thereto". The appellant's argument is not persuasive because the claims do not recite that the feedback signal is a combination of program material. Also, the appellant has argued that "the preferred embodiment starts with delayed feedback (delayed program and delayed talent) and undelayed talent signal to provided delayed program (without the delayed talent). The appellant's argument is not persuasive because as described above, Umemoto reference does disclose the delayed feedback signal (EC, which is inherently delayed when it reaches 14 from 13) and undelayed talent signal (RS) which then has an artificial

delay imposed to match the delay provided in EC. Signal EC has mixed with it the local speech ("program signal") entering 14 (Umemoto, col 3, lines 29-37).

On page 14, line 25 to page 15 line 2, the appellant has argued that "according to the examiner's interpretation RS is the talent signal which is most likely delayed in the transmission to the device, the feedback signal is identified as EC (which has no program content) which the examiner recognizes in in delayed". The appellant's argument is not persuasive because it is not clear to the examiner what the appellant means by "most likely delayed in the transmission to the device". Also, the claims do not recite that the feedback signal contains program content. In addition, as described above, the feedback signal is delayed when it is received at 14. As earlier remarked in regard to Umemoto, for example, the output of error junction 32 contains the local speech ("program signal") entering 14 plus the talent signal (RS), minus a delayed version of the talent signal (RS) from ADF 31.

On page 15, lines 3 to 19, the appellant has argued that "according to the examiner's configuration of Umemoto the undelayed feedback signal is coupled to the combining circuit to completely remove the feedback signal. There is not program signal or mix minus signal". There appellant's argument is not persuasive because the program signal is the speech generated by the user which goes into microphone 14. The echo from 13 is mixed with the locally generated speech and hence must be removed at 32 or it will be transmitted back to its source, forming a feedback loop. If there were no signal entering microphone 14 other than the signal RS emitted by speaker 13, then there would be no need for a microphone 14 at all, since the output of the error junction would, under proper convergence of ADF 31, be zero all the time. The point is that this is a speaker-phone system, and in order for the person speaking into 14 to easily hear the person on the other end, the other-end-person's voice is broadcast by speaker 13. Without the filter ADF and subtraction junction 32, the speech from the far end (RS) will be returned to that end and broadcast by another loudspeaker at that end, causing howlaround or acoustic feedback. With handset telephones, there is no EC; with speaker-phones, a method of removing EC from the near end speech entering microphone 14 is required, such as an adaptive filter ADF 31.

On page 15, lines 12 to 19, the appellant has argued that "assuming arguendo Umemoto's signal EC corresponds to the claimed feedback signal and RS corresponds to the claimed talent signals ad the examiner suggests, where is the program material in EC?". There appellant's argument is not persuasive because the claims do not recite "the program material in the feedback signal". Furthermore, as earlier remarked in regard to Umemoto, for example, the output of error junction 32 contains the local speech ("program signal") entering 14 plus the talent signal (RS), minus a delayed version of the talent signal (RS) from ADF 31. Thus there is indeed a program signal present, even though one is not called for by the claims.

On page 15, line 20 to page 16, line 8, the appellant has argued that "if the signal from the microphone 14 is considered to be the program signal than it is not delayed and the talent signal RS is not relatively undelayed". The appellant's argument is not persuasive because as described above, the feedback signal EC is inherently delayed when it reaches 14. Also, any local speech

entering microphone 14 also has a certain amount, however slight, of acoustic path delay, and is thus a delayed signal in the broadest sense.

On page 16, line 10 to 21, the appellant has argued that "the talent signal is not removed or canceled in 30 as claimed, it is passed". The appellant's argument is not persuasive because the feedback signal is canceled at 30 (see col. 4, lines 23 to 25).

On page 16, line 23 to page 17, line 11, the appellant has argued "although Kuo does describe a 'Correlation LMS Algorithm' there is not suggestion found in Kuo or Umemoto to use Kuo to provide the claimed correlation features". The appellant's argument is not persuasive because as described in above art rejection, concept of a correlation between two signals would have been obvious if not inherent.

On page 17, lines 16 to 26, the appellant has argued that "when Tanno is configured as suggested by the examiner, the feedback signal is undelayed (the examiner notes there is no variable delay) and the talent signal is delayed (by A1, S1 and the acoustic path from S1 to M2). The appellant's argument is not persuasive because when the feedback signal, which is output from S1, is received at M2, it is delayed. Also, the talent signal (a person speaks into M1) is relatively undelayed talent signal (as claimed).

On page 17, line 27 to page 18, line 15, the appellant has argued that "one of ordinary skill would not be motived look to acoustic feedback and cancellation technology such as Umemoto, Davidson and Tanno to achieve the various elements of the claimed invention". The appellant's argument is not persuasive because as described above, combining Umemoto, Davidson and Tanno would have been obvious to one of ordinary skill in the art. ***Furthermore, the claims, quite broad, are the measure of the invention.***

On page 18, line 16 to page 26, the appellant has argued that "in the broadcasting systems which the instant invention is intended to be used with, gains, once established, are usually very stable or at the most only very slowly changing. By contrast in acoustic systems the gains and delays are constantly required to change due to such factors as air movement, background noise, and even changes in physical location such as people moving about". The appellant's argument is not persuasive because the appellant's argument is beyond scope of the claims. No such distinction is recited in the claims.

On page 19, line 15 to page 20, line 2, the appellant has argued that " 'Umemoto's feedback signal is an acoustic feedback signal, that travels through the air from the mobile telephone speaker 13 to the microphone 14' and '(t) he feedback signal of the claims is not the same feedback signal which is addressed by the Umemoto invention' ". The appellant's argument is not persuasive because as described above, the feedback signal of Umemoto discloses the feedback signal ***as claimed***.

On page 20, lines 3 to 14, the appellant has argued that the examiner responded to the appellant's argument by only stating "the applicant's argument is not persuasive because it is not clear what the applicant means by 'not the same feedback signal' ". The examiner has stated that comment because the appellant has not explicitly defined how they were different.

On page 20, line 21 to page 21, line 7, the appellant has argued that "the rejections fail to state or explain why those meanings and limitations defined in the specification are not given to the various elements of the claims or why broader meanings and definitions are used". The appellant's argument is not persuasive because as described above the prior art of record discloses appellant's claimed invention.
Furthermore, it is customary to give terms their broadest reasonable interpretations. Many of Appellant's arguments concern details not claimed, or intended-use rather than structural (or in the case of method claims, step) limitations that can distinguish one signal from another.

For the above reasons, it is believed that the rejections should be sustained

For the above reasons, it is believed that the rejections should be sustained.

Respectfully submitted,

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PREFACE

Active noise control (ANC) is achieved by introducing a canceling "antinoise" wave through an appropriate array of secondary sources. These secondary sources are interconnected through an electronic system using a specific signal processing algorithm for the particular cancellation scheme. ANC is an effective way to attenuate noise that is very difficult and expensive to control using passive means. It has application to a wide variety of problems in manufacturing, industrial operations, and consumer products. Many industrial companies and universities are currently engaged in ANC research and development. This book is intended primarily as a reference source for researchers and engineers, with emphasis on signal processing and implementation; however, it can also serve as a text or reference for senior- or graduate-level electrical, mechanical, and acoustical engineering courses that are being offered at more and more colleges to present the necessary background for ANC.

The current wave of interest in ANC is due largely to the confluence of digital signal processing (DSP) hardware and adaptive signal processing algorithms, both of which have come into prominence within the last decade. This book aims to introduce the basic concepts of ANC from the standpoint of these two widely held perspectives.

We have chosen to take a broad definition of ANC as being concerned with the reduction of any kind of undesirable disturbance or noise, whether it is borne by electrical, acoustic, vibration, or any other kind of media. The key attribute of ANC that distinguishes it from older adaptive noise cancellation techniques is the presence of a transfer function from the adaptive filter output to the error-sensing node. In this book, we refer to this as the *secondary path*, which conveys the adaptive filter output signal through transducers and wave propagation to cancel the primary noise.

One of the authors originally introduced the notion of having a transfer function in the cancellation path of an adaptive filter in 1980 and suggested means for its compensation. Although that work was aimed at electrical problems and did not anticipate the more recent acoustic and vibration ANC

applications, it did lay some of the groundwork for grappling with the problem of a transfer function in the secondary path, which, if ignored, can lead to instability. In particular, a means for compensation was suggested, whereby the reference signal is filtered by an estimate of the secondary-path transfer function before correlating with the error feedback signal. Shortly after, in 1981, Burgess independently conceived this method within the context of acoustic duct noise cancellation and rigorously justified its usage by explicitly calculating the gradient of the instantaneous error. A similar notion was also independently introduced in 1981 by Widrow in the context of adaptive control systems. In that work, the term "filtered-X least-mean-square" (FXLMS) algorithm was coined, which has since gained widespread usage. This algorithm forms the cornerstone of ANC.

The emphasis of this book is on the practical aspects of ANC systems in terms of signal processing and DSP implementation. Thus, the principles of adaptive signal processing are combined with experimental results and practical issues, including the implementation of these structures and algorithms using C programs and assembly programs on DSP chips (TMS320C25 and TMS320C30 from Texas Instruments). In the way of theory, the book features concise derivations and analyses of commonly used adaptive structures and algorithms for ANC applications. A compilation of C and assembly programs is included in a floppy disk along with this book, and can be used to implement many ANC systems. This software (source code) is in a form ready to be used by students, researchers, and engineers.

We have tried to select application examples not to dazzle the cognoscenti, but rather to motivate graduate students and nonexperts in the field, who may not completely follow all the theoretical development, but could at least get a flavor of the idea by seeing some concrete examples and results of actual experiments. The amply cited literature in this book is replete with more advanced and more fully developed application examples, many of which are still in a state of evolution.

In matters of notation and style, we have spent much time and thought to make the most efficient and parsimonious mathematical representations. Moreover, we have sought to employ conventional usage as much as possible within the fields of mathematics, digital signal processing, adaptive signal processing, and active noise control, which are the main disciplines represented in this book. Some compromises were necessary, however, in order to reconcile differences between the various fields.

Discrete-time signals are assumed throughout with time index n in order to be compatible with most of the DSP literature. We use lowercase letters for time-domain signals and uppercase letters in the frequency domain, which is now standard practice. We have also adopted the preferred modern style of using bold lowercase for vectors and bold uppercase letters for matrices. (An occasional conflict occurs for representing vectors of frequency-domain quantities, in which case we defer to the uppercase.) When subscripts are employed

to enumerate a set of quantities, we try to use the suggestive notation of a lowercase running index and an uppercase index of the same letter for the total number of items, for instance, h_m, $m = 1, 2, \ldots, M$.

In the adaptive signal processing literature, the use of **w** for the adaptive weight vector is historic, starting with Widrow. In much of the literature, the number of adaptive weights is denoted by N. However, numbering the weights as w_n, $n = 0, 1, \ldots, N-1$ would be in conflict with the time index n. Therefore the weight index l was chosen and L adaptive weights are enumerated as w_l, $l = 0, 1, \ldots, L-1$. Another convention that we have adopted is to order the adaptive weight update term as $x(n)e(n)$, putting the reference signal term first. This makes the ordering consistent with the multiple-channel case in which the update takes the untransposable matrix-vector form $\mathbf{X}(n)\mathbf{e}(n)$, and is also consistent with the usual mathematical operator order, where the signal flow is from right to left (as opposed to signal flow diagrams, which go from left to right).

We have also tried to develop a uniform notation within the framework of ANC systems. In order to be consistent with the use of P for the primary path, we chose to represent the secondary path by S rather than using C as in some of the ANC literature, starting with Elliott. In representing the cancellation process, some devotees of ANC prefer to add the secondary path to the primary path. However, we prefer the convention of subtracting the secondary path from the primary path in order to be consistent with the vast body of adaptive filtering literature. The convention is rather arbitrary anyway, because one can always change the sign of the electrical signal feeding the secondary source to agree with whatever representation is desired.

In Chapter 1, we develop the basic philosophy of the ANC technique and explain why it is advantageous. Applications are cited in many fields. A general viewpoint of ANC is expressed that encompasses all types of noise media, such as air-acoustic, hydro-acoustic, and vibration.

In Chapter 2, we review adaptive filter theory and introduce commonly used algorithms that will carry over to ANC.

Chapter 3 covers broadband ANC. We discuss basic limitations due to coherence, derive the FXLMS algorithm, discuss feedback problems and solutions, and introduce the recursive filtered-U LMS algorithm. Practical system considerations are discussed throughout the development.

In Chapter 4, ANC techniques are specialized to the narrowband case, and the waveform synthesis method and adaptive notch filter are introduced.

Chapter 5 extends basic ANC techniques to multiple-channel algorithms and provides a unified and coherent treatment of the subject.

In Chapter 6, we first discuss classical nonadaptive feedback control to establish the background. Then, the concept of adaptive feedback ANC is developed from the standpoint of reference signal synthesis, thereby providing a link to the feedforward systems of Chapters 3 and 4. Finally, hybrid combinations of feedforward and feedback systems are considered.

Chapter 7 develops various on-line secondary-path modeling techniques. Attention is drawn to a fundamental bias problem, and several techniques for its solution are presented.

Chapter 8 introduces various special ANC algorithms and implementations such as the lattice ANC, frequency-domain ANC, recursive-least-squares (RLS) algorithm for ANC, subband ANC, and modal ANC.

Chapter 9 presents many examples of applications involving real and simulated experiments for air-acoustic and vibration problems.

As with any book attempting to capture the state of the art at a given time, there will necessarily be omissions, some intentional, some unintentional, that are necessitated by the rapidly evolving developments in this dynamic field of exciting theoretical and practical interest. We hope, at least, that this book will serve as a guide for what has already come and as an inspiration for what will follow.

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October 1995